



## INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

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<p><b>(54) Title: BURST-BY-BURST ADAPTIVE SINGLE-CARRIER DATA TRANSMISSION</b></p> <pre> graph TD     VE[Video Encoder] --&gt; PA[Packet Assembly]     PA --&gt; M1[Mapper]     M1 --&gt; FEC[n-class FEC Encoder]     FEC --&gt; M2[Mapper &amp; TDMA MPX]     M2 --&gt; QAM[1,2,4,6 bits/symbol QAM]     QAM --&gt; Channel[Channel]          Channel --&gt; QAMD[QAM Demodulator]     QAMD --&gt; M3[Mapper &amp; TDMA DEM]     M3 --&gt; FECD[n-class FEC Decoder]     FECD --&gt; M4[Mapper]     M4 --&gt; PD[Packet Disassembly]     PD --&gt; VE          FEEDBACK[Feedback information] --&gt; M3   </pre>			
<p><b>(57) Abstract</b></p> <p>The performance benefits of burst-by-burst adaptive modulation are studied, employing a higher-order modulation scheme, when the channel is favourable, in order to increase the system's bits per symbol capacity and conversely, invoking a more robust, lower order modulation scheme, when the channel exhibits inferior channel quality. It is shown that due to the described adaptive modem mode switching regime a seamless multimedia source-signal representation quality – such as video or audio quality – versus channel quality relationship can be established, resulting in a near-unimpaired multimedia source-signal quality right across the operating channel Signal-to-Noise Ratio (SNR) range. The main advantage of the described technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality – such as video or audio quality – by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This is achieved on a near-instantaneous or burst-by-burst adaptive basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile out-doors – or even hilly terrain – propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.</p>			

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## Title of the Invention

### Burst-by-burst Adaptive Single-carrier Data Transmission

#### 1 Background of the Invention

4 The invention relates to data transmission, more specifically to transmission in packets or bursts.

5 In contrast to the *burst-by-burst reconfigurable wideband* multimedia transceivers described in this doc-  
6 ument, the term *statically reconfigurable* found in this context in the literature refers to multimedia  
7 transceivers that cannot be near-instantaneously reconfigured. More explicitly, the previously proposed  
8 *statically reconfigurable* video transceivers were reconfigured on a long-term basis under the base sta-  
9 tion's control, invoking for example in the central cell region - where benign channel conditions prevail  
10 - a less robust, but high-throughput modulation mode, such as 4 bit/symbol Quadrature Amplitude Mod-  
11 ulation (16QAM), which was capable of transmitting a quadruple number of bits and hence ensured a  
12 better video quality. By contrast, a robust, but low-throughput modulation mode, such as 1 bit/symbol  
13 Binary Phase Shift Keying (BPSK) can be employed near the edge of the propagation cell, where hostile  
14 propagation conditions prevail. This prevented a premature hand-over at the cost of a reduced video  
15 quality.

16 The philosophy of the fixed, but programmable-rate proprietary video codecs and statically reconfigurable  
17 multi-mode video transceivers presented by Streit et al in References [1]-[4] was that irrespective of  
18 the video motion activity experienced, the specially designed video codecs generated a constant number  
19 of bits per video frame. For example, for videophony over the second-generation Global System of  
20 Mobile Communications known as the GSM system at 13 kbps and assuming a video scanning rate of  
21 10 frames/s, 1300 bits per video frame have to be generated. Specifically, two families of video codecs  
22 were designed, one refraining from using error-sensitive run-length coding techniques and exhibiting the  
23 highest possible error resilience and another, aiming for the highest possible compression ratio. This  
24 fixed-rate approach had the advantage of requiring no adaptive feedback controlled bitrate fluctuation  
25 smoothing buffering and hence exhibited no objectionable video latency or delay. Furthermore, these  
26 video codecs were amenable to video telephony over fixed-rate second-generation mobile radio systems,  
27 such as the GSM.

28 The fixed bitrate of the above proprietary video codecs is in contrast to existing standard video codecs,  
29 such as the Motion Pictures Expert Group codecs known as MPEG1 and MPEG2 or the ITU's H.263  
30 codec, where the time-variant video motion activity and the variable-length coding techniques employed

31 result in a time-variant bitrate fluctuation and a near-constant perceptual video quality. This time-variant  
32 bitrate fluctuation can be mitigated by employing adaptive feed-back controlled buffering, which poten-  
33 tially increases the latency or delay of the codec and hence it is often objectionable for example in inter-  
34 active videophony. The schemes presented by Streit et al in References [1]-[4] result in slightly variable  
35 video quality at a constant bitrate, while refraining from employing buffering, which again, would result  
36 in latency in interactive videophony. A range of techniques, which can be invoked, in order to render the  
37 family of variable-length coded, highly bandwidth-efficient, but potentially error-sensitive class of stan-  
38 dard video codecs, such as the H.263 arrangement, amenable to error-resilient, low-latency interactive  
39 wireless multimode videophony was summarised in [5]. The adaptive video rate control and packetisa-  
40 tion algorithm of [5] generates the required number of bits for the burst-by-burst adaptive transceiver,  
41 depending on the capacity of the current packet, as determined by the current modem mode. Fur-  
42 ther error-resilient H.263-based schemes were contrived for example by Färber, Steinbach and Girod  
43 at Erlangen University [6], while Sadka, Eryurtlu and Kondoz [7] from Surrey University proposed a  
44 range of improvements to the H.263 scheme. Following the above portrayal of the prior art in both video  
45 compression and statically reconfigurable narrowband modulation, let us now consider the philosophy of  
46 wideband burst-by-burst adaptive quadrature amplitude modulation (AQAM) in more depth.

47 In burst-by-burst adaptive modulation a higher-order modulation scheme is invoked, when the channel  
48 is favourable, in order to increase the system's bits per symbol capacity and conversely, a more robust  
49 lower order modulation scheme is employed, when the channel exhibits inferior channel quality, in order  
50 to improve the mean Bit Error Ratio (BER) performance. A practical scenario, where adaptive modula-  
51 tion can be applied is, when a reliable, low-delay feedback path is created between the transmitter and  
52 receiver, for example by superimposing the estimated channel quality perceived by the receiver on the  
53 reverse-direction messages of a duplex interactive channel. The transmitter then adjusts its modem mode  
54 according to this perceived channel quality.

55 Recent developments in adaptive modulation over a narrow-band channel environment have been pi-  
56oneered by Webb and Steele [9], where the modulation adaptation was utilized in a Digital European  
57 Cordless Telephone - like (DECT) system. The concept of variable rate adaptive modulation was also  
58 advanced by Sampei *et al* [12, 17], showing promising advantages, when compared to fixed modula-  
59 tion in terms of spectral efficiency, BER performance and robustness against channel delay spread. In  
60 another paper, the numerical upper bound performance of adaptive modulation in a slow Rayleigh flat-  
61 fading channel was evaluated by Torrance *et al*[10] and subsequently, the optimization of the switching  
62 threshold levels using Powell minimization was used in order to achieve a targeted performance [11, 18].  
63 In addition, adaptive modulation was also studied in conjunction with channel coding and power control

64 techniques by Matsuoka *et al* [12] as well as Goldsmith *et al.* [13]-[15].  
65 In the narrow-band channel environment, the quality of the channel was determined by the short term  
66 Signal to Noise Ratio (SNR) of the received burst, which was then used as a criterion in order to choose  
67 the appropriate modulation mode for the transmitter, based on a list of switching threshold levels,  $l_n$  [9,  
68 10]. However, in a wideband environment, this criterion is not an accurate measure for judging the quality  
69 of the channel, where the existence of multi-path components produces not only power attenuation of the  
70 transmission burst, but also intersymbol interference. Subsequently, a new criterion has to be defined to  
71 estimate the wideband channel quality in order to choose the appropriate modulation scheme.

## 72 2 Summary of the Invention

73 Particular and preferred aspects of the invention are set out in the accompanying independent and depen-  
74 dent claims. Features of the dependent claims may be combined with those of the independent claims as  
75 appropriate and in combinations other than those explicitly set out in the claims.

76 The performance benefits of burst-by-burst adaptive modulation are described, employing a higher-order  
77 modulation scheme, when the channel is favourable, in order to increase the system's bits per symbol  
78 capacity and conversely, invoking a more robust, lower order modulation scheme, when the channel  
79 exhibits inferior channel quality. It is shown that due to the described adaptive modem mode switch-  
80 ing regime a seamless multimedia source-signal representation quality - such as video or audio quality -  
81 versus channel quality relationship can be established, resulting in a near-unimpaired multimedia source-  
82 signal quality right across the operating channel Signal-to-Noise Ratio (SNR) range. The main advan-  
83 tage of the described technique is that irrespective of the prevailing channel conditions, the transceiver  
84 achieves always the best possible source-signal representation quality - such as video or audio quality - by  
85 automatically adjusting the achievable bitrate and the associated multimedia source-signal representation  
86 quality in order to match the channel quality experienced. This can be achieved on a near-instantaneous or  
87 burst-by-burst adaptive basis under given propagation conditions in order to cater for the effects of path-  
88 loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when a mobile is  
89 roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order,  
90 low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate,  
91 high source-signal representation quality modes are employed.

### 92 3 Brief Description of the Drawings

93 For a better understanding of the invention and to show how the same may be carried into effect reference  
94 is now made by way of example to the accompanying drawings, in which:

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119 shown with a realistic one TDMA frame delay between channel estimation and mode  
120 switching, and also as a zero delay version for indicating the upper bound. The channel  
121 parameters were defined in Table 1. . . . .

122 10 Decoded video quality (PSNR) versus channel SNR comparison of the four fixed mod-  
123 uation modes (BPSK, 4QAM, 16QAM, 64QAM) with 5% transmission FER switching  
124 and that of the adaptive burst-by-burst modem (AQAM). AQAM is shown with a realis-  
125 tic one TDMA frame delay between channel estimation and mode switching, and a zero  
126 delay version for indicating the upper bound. The channel parameters were defined in  
127 Table 1. . . . .

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129 dem (AQAM). AQAM is shown with a realistic one TDMA frame delay between channel  
130 estimation and mode switching, and a zero delay version indicating the upper bound. Re-  
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135 of the adaptive burst-by-burst modem (AQAM). AQAM is shown with a realistic one  
136 TDMA frame delay between channel estimation and mode switching, and a zero delay  
137 version indicating the upper bound. The channel parameters were defined in Table 1. . . .

138 13 Transmission FER (or packet loss ratio) versus Channel SNR comparison of the fixed  
139 BPSK modulation mode and the adaptive burst-by-burst modem (AQAM) for the three  
140 sets of switching thresholds described in Table 4. AQAM is shown with a realistic one  
141 TDMA frame delay between channel estimation and mode switching. The channel pa-  
142 rameters were defined in Table 1. . . . .

143 14 Video bitrate versus channel SNR comparison for the adaptive burst-by-burst modem  
144 (AQAM) with a realistic one TDMA frame delay between channel estimation and mode  
145 switching for the three sets of switching thresholds as described in Table 4. The channel  
146 parameters were defined in Table 1. . . . .

## 147 4 Detailed Description

### 148 4.1 General Introduction to Adaptive Modem Mode Signalling Scenarios

149 AQAM transmission parameter adaptation is an action of the transmitter in response to time-varying  
150 channel conditions. It is only suitable for duplex communication between two stations, since the trans-  
151 mission parameter adaptation relies on some form of channel estimation and signalling. In order to  
152 efficiently react to the changes in channel quality, the following steps have to be taken:

- 153 • *Channel quality estimation:* In order to appropriately select the transmission parameters to be  
154 employed for the next transmission, a reliable prediction of the channel quality during the next  
155 active transmit timeslot is necessary.
- 156 • *Choice of the appropriate parameters for the next transmission:* Based on the prediction of the  
157 expected channel conditions during the next timeslot, the transmitter has to select the appropriate  
158 modulation schemes for the subcarriers.
- 159 • *Signalling or blind detection of the employed parameters:* The receiver has to be informed, as  
160 to which set of demodulator parameters to employ for the received packet. This information can  
161 either be conveyed within the packet, at the cost of loss of useful data bandwidth, or the receiver  
162 can attempt to estimate the parameters employed at the transmitter by means of blind detection  
163 mechanisms.

164 Depending on the channel characteristics, these operations can be performed at either of the duplex  
165 stations, as shown in Figure 1. If the channel is reciprocal, then the channel quality estimation for each  
166 link can be extracted from the reverse link, and we refer to this regime as open-loop adaptation. In this  
167 case, the transmitter needs to communicate the transmission parameter set to the receiver (Figure 1(a)),  
168 or the receiver can attempt blind detection of the transmission parameters employed (Figure 1(c)).  
169 If the channel is not reciprocal, then the channel quality estimation has to be performed at the receiver  
170 of the link. In this case, the channel quality measure or the set of requested transmission parameters is  
171 communicated to the transmitter in the reverse link (Figure 1(b)). This mode is referred to as closed-loop  
172 adaptation.

### 173 4.2 A Specific Embodiment of a Video Transceiver

174 The schematic of the whole system is depicted in Figure 2. In the described system the wideband channel-  
175 induced degradation is combated not only by the employment of adaptive modulation but also by equal-  
176 ization, where the equalization process will eliminate most of the intersymbol interference based on a

177 Channel Impulse Response (CIR) estimate derived using the channel sounding midamble and conse-  
178 quently, the signal to noise and residual interference ratio at the output of the equalizer is calculated.  
179 We note, however that the above adaptive methodology can also be extended to employing burst-by-  
180 burst adaptive channel coding associated with different-strength error correction codecs in the different  
181 transceiver modes of operation.

### 182 4.3 Channel quality metrics

183 The most reliable channel quality estimate is the bit error rate (BER), since it reflects the channel quality,  
184 irrespective of the source or the nature of the quality degradation.

185 Firstly, the BER can be estimated with a certain granularity or accuracy, provided that the system entails  
186 a channel decoder or - synonymously - Forward Error Correction (FEC) decoder employing algebraic  
187 decoding.

188 Secondly, if the system contains a soft-in-soft-out (SISO) channel decoder, the BER can be estimated  
189 with the aid of the Logarithmic Likelihood Ratio (LLR), evaluated either at the input or the output of the  
190 channel decoder. A particularly attractive way of invoking LLRs is employing powerful turbo codecs,  
191 which provide a reliable indication of the confidence associated with a particular bit decision in the  
192 context of LLRs. The LLR is defined as the ratio of the probabilities of a specific bit being binary zero  
193 or one. Again, this measure can be evaluated at both the input and the output of the turbo channel codecs  
194 and both of them can be used for channel quality estimation.

195 Thirdly, in the event that no channel encoder / decoder (codec) is used in the system, the channel quality  
196 expressed in terms of the BER can be estimated with the aid of the mean-squared error (MSE) at the  
197 output of the channel equaliser or the closely related metric, the Pseudo-Signal-to-noise-ratio (Pseudo-  
198 SNR). The MSE or pseudo-SNR at the output of the channel equaliser have the important advantage  
199 that they are capable of quantifying the severity of the inter-symbol-interference (ISI) and/or Co-channel  
200 Interference experienced, in other words quantifying the Signal to Interference plus Noise Ratio (SINR).

#### 201 4.3.1 Pseudo-SNR Embodiment

202 A specific embodiment based on the above-mentioned pseudo-SNR is now described in more depth.  
203 Employing the pseudo-SNR has the advantage that it is generally applicable, regardless of whether or  
204 not a channel codec is present.

We found that the residual channel-induced inter-symbol-interference (ISI) at the output of the decision  
feedback equaliser (DFE) is near-Gaussian distributed and that if there are no decision feedback errors,

the pseudo-SNR at the output of the DFE,  $\gamma_{DFE}$  can be calculated as [8]:

$$\begin{aligned}\gamma_{DFE} &= \frac{\text{Wanted Signal Power}}{\text{Residual ISI Power + Effective Noise Power}} \\ &= \frac{E\left[|S_k \sum_{m=0}^{N_f-1} C_m h_m|^2\right]}{\sum_{q=-(N_f-1)}^{-1} E\left[|f_q S_{k-q}|^2\right] + N_o \sum_{m=0}^{N_f-1} |C_m|^2}.\end{aligned}\quad (1)$$

where  $C_m$  and  $h_m$  denotes the DFE's feed-forward coefficients and the channel impulse response, respectively. The transmitted signal and the noise spectral density is represented by  $S_k$  and  $N_o$ . Lastly, the number of DFE feed-forward coefficients is denoted by  $N_f$ . By utilizing the pseudo-SNR at the output of the equalizer, we are ensuring that the system performance is optimised by employing equalization and adaptive quadrature amplitude modulation (AQAM) in a wideband environment according to the following switching regime:

$$\text{Modulation Mode} = \begin{cases} \text{BPSK} & \text{if } \gamma_{DFE} < f_1 \\ \text{4QAM} & \text{if } f_1 < \gamma_{DFE} < f_2 \\ \text{16QAM} & \text{if } f_2 < \gamma_{DFE} < f_3 \\ \text{64QAM} & \text{if } \gamma_{DFE} > f_3, \end{cases} \quad (2)$$

205 where  $f_n, n = 1 \dots 3$  are the pseudo-SNR thresholds levels, which are set according to the system's integrity  
206 requirements.

207 In contrast to the narrowband, statically reconfigured multimode systems of [1]-[4] constituting the state-  
208 of-the-art, the present embodiment invokes wideband, near-instantaneously reconfigured or burst-by-  
209 burst adaptive channel-equalised modulation, in order to achieve the best possible multimedia source-  
210 signal representation quality - for example video quality - when transmitting over arbitrarily time-variant  
211 channels, exhibiting short-term and/or long-term channel quality variations. These variations can be  
212 due to the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Further-  
213 more, when the mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment,  
214 typically low-order, low-rate modem modes are invoked, while in benign indoor environments predomi-  
215 nantly the high-rate, high video quality modes are employed.

216 It is an important element of the system that when the binary BCH channel codes or FEC codes protect-  
217 ing the video stream are overwhelmed by the plethora of transmission errors, the embodiment refrains  
218 from decoding the video packet in order to prevent error propagation through the reconstructed frame  
219 buffer [5]. Instead, these corrupted packets are dropped and the reconstructed frame buffer will not be  
220 updated, until the next packet replenishing the specific video frame area arrives. The associated video

Parameter	Value
Carrier Frequency	1.9GHz
Vehicular Speed	30mph
Doppler frequency	85Hz
Normalised Doppler frequency	$3.27 \times 10^{-5}$
Channel type	COST 207 Typical Urban (see Figure 3)
Number of paths in channel	4
Data modulation	Adaptive QAM (BPSK, 4-QAM, 16-QAM, 64-QAM)
Receiver type	Decision Feedback Equalizer Number of Forward Filter Taps = 35 Number of Backward Filter Taps = 7

Table 1: Modulation and channel parameters

221 performance degradation is fairly minor for packet dropping or frame error rates (FER) below about 5%.  
 222 These packet dropping events are signalled to the remote decoder by superimposing a strongly protected  
 223 one-bit packet acknowledgement flag on the reverse-direction packet, as outlined in [5]. In the embod-  
 224 iment we also invoked the adaptive rate control and packetisation algorithm of [5], supporting constant  
 225 Baud-rate operation.  
 226 As a specific example of the burst-by-burst adaptive multimedia system we used 176x144 pixel so-called  
 227 QCIF-resolution, 30 frames/s video sequences encoded at bitrates resulting in high perceptual video  
 228 quality, in order to demonstrate the performance advantages of the video transceiver. Table 1 shows the  
 229 modulation- and channel parameters employed, noting again that the associated principles are applicable  
 230 in the context of a whole range of other system parameters. The COST207 four-path typical urban (TU)  
 231 channel model was used in quantifying the associated system performance and its impulse response  
 232 is portrayed in Figure 3. As an example, we used the Pan-European FRAMES proposal as the basis  
 233 for our wideband transmission system, the frame structure of which is shown in Figure 4. Employing  
 234 the FRAMES Mode A1 (FMA1) so-called non-spread data burst mode required a system bandwidth of  
 235 3.9MHz, when assuming a modulation excess bandwidth of 50%. A range of other system parameters  
 236 are shown in Table 2.  
 237 The specific example of the video transceiver - which is used to demonstrate the advantages of the system  
 238 concept - is based on the H.263 video codec. The video coded bitstream was protected by binary Bose-

Features	Value
Multiple access	TDMA
No. of Slots/Frame	16
TDMA frame length	4.615ms
TDMA slot length	288 $\mu$ s
Data Symbols/TDMA slot	684
User Data Symbol Rate (KBd)	148.2
System Data Symbol Rate (MBd)	2.37
Symbols/TDMA slot	750
User Symbol Rate (KBd)	162.5
System Symbol Rate (MBd)	2.6
System Bandwidth (MHz)	3.9
Eff. User Bandwidth (kHz)	244

Table 2: Generic system features of the reconfigurable multi-mode video transceiver, using the non-spread data burst mode of the FRAMES proposal shown in Figure 4.

239 Chaudhuri-Hocquenghem (BCH) coding combined with an intelligent burst-by-burst adaptive wideband  
 240 multi-mode Quadrature Amplitude Modulation (QAM) modem, which can be configured either under  
 241 network control or under transceiver control to operate as a 1, 2, 4 and 6 bits/symbol scheme, while  
 242 maintaining a constant signalling rate. This allowed us to support an increased throughput expressed  
 243 in terms of the average number of bits per symbol, when the instantaneous channel quality was high,  
 244 leading ultimately to an increased video quality in a constant bandwidth.

245 The transmitted bitrate for all four modes of operation is shown in Table 3. The unprotected bitrate  
 246 before approximately half-rate BCH coding is also shown in Table 3. The actual useful bitrate available  
 247 for video is slightly less, than the unprotected bitrate due to the required strongly protected packet ac-  
 248 knowledgegement information and packetisation information. The effective video bitrate is also shown in  
 249 Table 3.

#### 250 4.4 Burst-by-Burst Adaptive Videophone Performance

251 The described burst-by-burst adaptive modem maximizes the system capacity available by using the  
 252 most appropriate modulation mode for the current instantaneous channel conditions. We found that the  
 253 pseudo-SNR at the output of the channel equaliser was an adequate channel quality measure in our burst-

Features	Multi-rate System			
Mode	BPSK	4QAM	16QAM	64QAM
Bits/Symbol	1	2	4	6
FEC	Near Half-rate BCH			
Transmission bitrate (kbit/s)	148.2	296.4	592.8	889.3
Unprotected bitrate (kbit/s)	75.8	151.7	303.4	456.1
Effective Video-rate (kbit/s)	67.0	141.7	292.1	446.4
Video fr. rate (Hz)	30			

Table 3: Operational-mode specific transceiver parameters

254 by-burst adaptive wide-band modem. Figure 5 demonstrates how the burst-by-burst adaptive modem  
 255 changes its modulation modes every transmission burst, ie every 4.615 ms, based on the fluctuating  
 256 pseudo-SNR. The right-hand-side vertical axis indicates the associated number of bits per symbol.  
 257 By changing to more robust modulation schemes automatically, when the channel quality is reduced  
 258 allows the packet loss ratio, or synonymously, the FER, to be reduced, which results in increased per-  
 259 ceived video quality. In order to judge the benefits of burst-by-burst adaptive modulation we considered  
 260 two scenarios. In the first scheme the adaptive modem always chose the perfectly estimated AQAM  
 261 modulation mode, in order to provide a maximum upper bound performance. In the second scenario  
 262 the modulation mode was based upon the perfectly estimated AQAM modulation mode for the previous  
 263 burst, which corresponded to a delay of one Time Division Multiple Access (TDMA) frame duration  
 264 of 4.615ms. This second scenario represents a practical burst-by-burst adaptive modem, where the one-  
 265 frame channel quality estimation latency is due to superimposing the receiver's perceived channel quality  
 266 on a reverse-direction packet, for informing the transmitter concerning the best mode to be used.  
 267 The probability of the adaptive modem using each modulation mode for a particular average channel  
 268 SNR is portrayed in Figure 6 in terms of the associated modem mode probability density functions  
 269 (PDFs). It can be seen at high channel SNRs that the modem mainly uses the 64QAM modulation mode,  
 270 while at low channel SNRs the BPSK mode is the most prevalent one.  
 271 Figure 7 shows the transmission FER (or packet loss ratio) versus channel SNR for the 1, 2, 4 and 6  
 272 bit/symbol fixed modulation schemes, as well as for the ideal and for the one-frame delayed realistic

273 scenarios using the burst-by-burst adaptive QAM (AQAM) modem. In the ideal - ie zero-delay - AQAM  
274 scenario, where the modulation mode estimation is assumed to be available instantaneously, the trans-  
275 mission FER is zero at high channel SNRs even though 64QAM is used predominantly, while at low  
276 SNRs it exhibits a similar FER behaviour to fixed BPSK modulation, since this is the most often used  
277 mode. More explicitly, at high SNRs the adaptive modem chooses the most suitable AQAM mode and  
278 hence no packets are lost. However, at low SNRs the adaptive modem opts for using BPSK, even when  
279 the channel is so hostile that the packets are lost. Hence the BPSK and ideal - ie zero-delay - AQAM  
280 results are very similar at low channel SNRs. However, when the modulation mode estimation is delayed  
281 by one TDMA frame - representing a realistic, practical AQAM modem - then the transmission FER is  
282 no longer zero at high channel SNRs, since the delay results in a non-optimum modulation mode selec-  
283 tion, which can result in the corresponding video packet being lost. At high channel SNRs the FER of  
284 the realistic, one-frame delay AQAM modem is similar to that of the fixed 64QAM modem mode. By  
285 contrast, at low channel SNRs its FER performance is similar to that of the fixed BPSK modem mode.  
286 However, at medium channel SNRs the transmission FER is almost constant at about 3% for the realistic  
287 AQAM modem. This is more clearly demonstrated on a logarithmic scale in Figure 8.

288 Explicitly, the ideal and realistic AQAM modems are compared to a fixed modulation based, statically  
289 re-configured multi-mode system with switching at 5% transmission FER in Figure 8. The statically  
290 reconfigured modem was invoked here as a benchmark, in order to indicate, how a system would  
291 perform, which cannot act on the basis of the near-instantaneously varying channel quality. As it can  
292 be inferred from Figure 8, such a statically reconfigured transceiver switches its mode of operation from  
293 a lower-order modem mode, such as for example BPSK to a higher-order mode, such as 4QAM, when  
294 the channel quality has improved sufficiently for the 4QAM mode's FER to become lower than 5 %  
295 upon reconfiguring the transceiver in this 4QAM mode. Again - as seen in Figure 7 earlier on a non-  
296 logarithmic scale - Figure 8 clearly shows that the ideal AQAM modem has a similar FER performance  
297 to the fixed rate BPSK modem. Additionally, it indicates that the realistic AQAM modem has a similar  
298 FER performance to the BPSK modem at low SNRs, yielding a near-constant 3% FER at medium SNRs  
299 and a FER similar to that of the fixed 64QAM modem at high channel SNRs.

300 A comparison of the effective video bitrate versus channel SNR is shown in Figure 9 for the four fixed  
301 modulation schemes, and the ideal and realistic AQAM modems. The effective video bitrate is the  
302 average bitrate provided by all the successfully transmitted video packets. It should be noted that the  
303 realistic AQAM modem has a slightly lower throughput, since sometimes the incorrect modulation mode  
304 is chosen, which may result in packet loss. At very low channel SNRs the throughput bitrate converges  
305 to that of the fixed BPSK mode, since the AQAM modem is almost always in the BPSK mode at these

306 SNRs, as demonstrated in Figure 6.

307 Having shown the effect of the burst-by-burst adaptive modem on the transmission FER and effective  
308 bitrate, let us now demonstrate these effects on the decoded video quality, measured in terms of the Peak  
309 Signal-to-Noise Ratio (PSNR). Figure 10 shows the decoded video quality in terms of PSNR versus  
310 channel SNR for both the ideal and realistic adaptive modem, and for the four modes of the statically  
311 configured multi-mode modem. It can be seen that - as expected - the ideal adaptive modem, which  
312 always selects the perfect modulation modes, has a better or similar video quality for the whole range  
313 of channel SNRs. For the statically configured multi-mode scheme the video quality degrades, when  
314 the system switches from a higher-order to a lower-order modulation mode. The ideal adaptive modem  
315 however smoothes out the sudden loss of video quality, as the channel degrades. The non-ideal adaptive  
316 modem has a slightly lower video quality performance, than the ideal adaptive modem, especially at  
317 medium SNRs, since it sometimes selects the incorrect modulation mode due to the estimation delay.  
318 This can inflict video packet loss and/or a reduction of the effective video bitrate, which in turn reduces  
319 the video quality.

320 The difference between the ideal burst-by-adaptive modem, using ideal channel estimation and the non-  
321 ideal, realistic burst-by-burst adaptive modem, employing a non-ideal delayed channel estimation can be  
322 seen more clearly in Figure 11 for a range of video sequences. Observe that at high and low channel  
323 SNRs the video quality performance is similar for the ideal and non-ideal adaptive modems. This is  
324 because the channel estimation delay has little effect, since at low or high channel SNRs the adaptive  
325 modem is in either BPSK or 64QAM mode for the majority of the time. More explicitly, the channel  
326 quality of a transmission frame is almost always the same as that of the next, and hence the delay has  
327 little effect at low and high SNRs.

328 The video quality versus channel quality trade-offs can be more explicitly observed in Figure 12. This  
329 figure portrays the decoded video quality in PSNR versus the packet loss ratio or transmission FER.  
330 The ideal and practical adaptive modem performance is plotted against that of the four fixed modulation  
331 schemes in the figure. It can be seen that the adaptive modems' video quality degrades from that  
332 achieved by the error-free 64QAM modem towards the BPSK modem performance as the packet loss  
333 ratio increases. The practical adaptive modems' near constant FER performance of 3% at medium SNRs  
334 can be clearly seen in the figure, which is associated with the reduced PSNRs of the various modem  
335 modes, while having only minor channel error-induced impairments.

	BPSK	4QAM	16QAM	64QAM
Standard	<10dB	≥10dB	≥18dB	≥24dB
Conservative	<13dB	≥13dB	≥20dB	≥26dB
Aggressive	<9dB	≥9dB	≥17dB	≥23dB

Table 4: SINR estimate at output of the equaliser required for each modulation mode in Burst-by-Burst Adaptive modem, ie. switching thresholds

#### 336 4.5 Switching Thresholds

337 The burst-by-burst adaptive modem changes its modulation modes based on the fluctuating channel con-  
 338 ditions expressed in terms of the SNR at the equaliser's output. The set of switching thresholds used in  
 339 all the previous graphs is the "Standard" set shown in Table 4, which was determined on the basis of the  
 340 required channel SINR for maintaining the specific target video FER.

341 In order to investigate the effect of different sets of switching thresholds, we defined two new sets  
 342 of thresholds, a more conservative set, and a more aggressive set, employing less robust, but more  
 343 bandwidth-efficient modem modes at lower SNRs. The more conservative switching thresholds reduced  
 344 the transmission FER at the expense of a lower effective video bitrate. The more aggressive set of thresh-  
 345 olds increased the effective video bitrate at the expense of a higher transmission FER.

346 The transmission FER performance of the realistic burst-by-burst adaptive modem, which has a one  
 347 TDMA frame delay between channel quality estimation and mode switching is shown in Figure 13 for  
 348 the three sets of switching thresholds of Table 4. It can be seen that the more conservative switching  
 349 thresholds reduce the transmission FER from about 3% to about 1% for medium channel SNRs. The  
 350 more aggressive switching thresholds increase the transmission FER from about 3% to 4-5%. However,  
 351 since FERs below 5% are not objectionable in video quality terms, this FER increase is an acceptable  
 352 compromise for a higher effective video bitrate. The effective video bitrate for the realistic adaptive  
 353 modem with the three sets of switching thresholds is shown in Figure 14. The more conservative set  
 354 of switching thresholds reduces the effective video bitrate but also reduces the transmission FER. The  
 355 aggressive switching thresholds, increase the effective video bitrate, but also increase the transmission  
 356 FER. Therefore the optimal switching thresholds should be set such that the transmission FER is deemed  
 357 acceptable is the range of channel SNRs considered.

## 358 5 Summary

359 The above-described burst-by-burst adaptive multimedia transceiver concept exhibits substantial advan-  
360 tages in comparison to conventional fixed-mode or statically reconfigurable transceivers, which was sub-  
361 stantiated in the context of a specific embodiment of the advocated system concept, namely with the aid  
362 of a burst-by-burst adaptive video transceiver.

363 Specifically, the main advantage of the described burst-by-burst adaptive transceiver technique is that ir-  
364 respective of the prevailing channel conditions, the transceiver achieves always the best possible source-  
365 signal representation quality - such as video, speech or audio quality - by automatically adjusting the  
366 achievable bitrate and the associated multimedia source-signal representation quality in order to match  
367 the channel quality experienced. This is achieved on a near-instantaneous or burst-by-burst adaptive  
368 basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-  
369 fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile  
370 out-doors - or even hilly terrain - propagation environment, typically low-order, low-rate modem modes  
371 are invoked, while in benign indoor environments predominantly the high-rate, high source-signal repre-  
372 sentation quality modes are employed.

373 The described system embodiment has the following features:

- 374 1. A reliable instantaneous channel quality metric is employed, in order to appropriately configure  
375 the AQAM modem for maintaining the required target BER and the associated source signal rep-  
376 resentation quality. The range of potential channel quality metrics entails the pseudo-SNR, SINR,  
377 BER and its LLR-based channel estimates.
- 378 2. The perceived channel quality determines the number of bits that can be transmitted in a given  
379 transmitted packet or burst, which in turn predetermines the number of bits to be generated by the  
380 associated multimedia source codec, such as for example the associated video, audio, speech or  
381 handwriting codec. Hence the multimedia source codec has to be capable of adjusting the number  
382 of bits generated under the instruction of the burst-by-burst adaptive transceiver.
- 383 3. The transmitter mode requested by the receiver, in order to achieve the target performance has to  
384 be signalled by the receiver to the remote transmitter. Another scenario is, where the uplink and  
385 downlink channel quality is sufficiently similar for allowing the receiver to judge, what transmis-  
386 sion mode the associated transmitter should use, in order for its transmitted signal to maintain the  
387 required transmission integrity. Lastly, the mode of operation used by the transmitter can also be  
388 detected using blind detection techniques, for example in conjunction with the associated channel

389 decoder.

390 In the studied example of the system embodiment we have characterised a wideband burst-by-burst adaptive multimedia transceiver, which employed the pseudo-SNR at the output of the channel equaliser as 391 the quality measure for controlling the AQAM modem modes. Whilst in reference [16] the through- 392 put upper-bound of such an AQAM modem was analysed, in this document a practical multimedia 393 transceiver concept was described and the achievable performance gains due to employing the described 394 wideband bursts-by-burst adaptive modem were quantified. An adaptive packetiser was used in conjunc- 395 tion with the adaptive modem, which continually adjusted the video codec's target bitrate, in order to 396 exploit the instantaneous bitrate provided by the adaptive modem.

397 In the example the delay between the channel estimation and modulation mode switching was shown 398 to have a considerable effect on the performance achieved by the adaptive modem. This performance 399 penalty can be mitigated by reducing the modem mode switching latency, for example by employing 400 adjacent slots for the uplink and downlink of a TDD system. However, at lower vehicular speeds 401 the effects of AQAM mode switching latency are less crucial and the practical adaptive modem can 402 achieve a performance that is close to that of the ideal adaptive modem exhibiting no switching latency, 403 that we used as an upper-bound benchmark. We have also demonstrated, how the transmission FER 404 performance is affected by changing the switching thresholds. Therefore the system can be tuned to the 405 required FER performance using appropriate switching thresholds.

406 It will be appreciated that although a particular embodiment of the invention has been described, many 407 modifications / additions and / or substitutions may be made within the spirit and scope of the present 408 invention.

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CLAIMS

1. A receiver unit comprising:
  - a burst-by-burst adaptive equalizer having an input for receiving data bursts from a communication channel, each burst containing a number of bits per symbol;
  - a computational unit for computing a received signal quality metric related to a bit error rate experienced during transmission over the communication channel;
- 10 an output for relaying the signal quality metric, conveying signal quality as perceived by the receiver unit, for use in determining a configuration for subsequent transmission bursts.
2. A receiver unit according to claim 1, wherein the received signal quality metric is evaluated from an interference parameter.
- 15 3. A receiver unit according to claim 2, wherein the signal quality metric is evaluated using channel impulse response estimates derived from a training sequence embedded in each transmission burst.
- 20 4. A receiver unit according to claim 2, wherein the interference parameter is a measure of co-channel interference.
- 25 5. A receiver unit according to claim 2, wherein the interference parameter is a measure of inter-symbol interference.

6. A receiver unit according to claim 1, wherein the signal quality metric is evaluated according to the formula:

$$\begin{aligned}
 \gamma_{dfe} &= \frac{\text{Wanted Signal Power}}{\text{Residual ISI Power + Effective Noise Power}} \\
 5 &= \frac{E \left[ |S_k \sum_{m=0}^{N_f-1} C_m h_m|^2 \right]}{\sum_{q=-(N_f-1)}^{N_f-1} E \left[ |f_q S_{k-q}|^2 \right] + N_0 \sum_{m=0}^{N_f-1} |C_m|^2}
 \end{aligned}$$

10 where  $\gamma_{dfe}$  is pseudo-SNR at the output of the equalizer,  $C_m$  and  $H_m$  denote feed-forward coefficients and the channel impulse response respectively,  $S_k$  and  $N_0$  are transmitted signal and noise spectral density respectively, and  $N_f$  is the number of feed-forward coefficients.

15 7. A receiver unit according to claim 1, wherein the received signal quality metric is evaluated from the bit error rate.

8. A receiver unit according to claim 7, wherein the bit error rate is estimated by an algebraic channel decoder.

20 9. A receiver unit according to claim 7, further comprising a channel decoder arranged to receive the data bursts from the adaptive equalizer, and wherein the bit error rate is estimated from a calculation of a logarithmic likelihood ratio, thereby to provide a reliable estimator of all possible channel impairments.

25 10. A receiver unit according to claim 9, wherein the logarithmic likelihood ratio is calculated at the input of the channel decoder.

30 11. A receiver unit according to claim 9, wherein the logarithmic likelihood ratio is calculated at the output of the channel decoder.

12. A receiver unit according to any one of the preceding claims, wherein the configuration defines the number of bits per symbol in each transmission burst, which is varied according to the signal quality metric computed from a previous transmission burst, as supplied by the output.

5

13. A receiver unit according to any one of the preceding claims, wherein the output is arranged to relay the signal quality metric, representative of signal quality as perceived by the receiver unit, through the communication channel for configuration of the remote transmitter for subsequent transmission bursts, thereby to provide  
10 closed-loop feedback.

10

14. A receiver unit according to any one of claims 1 to 12, wherein the output is arranged to relay the signal quality metric, representative of signal quality as perceived by the receiver unit, to a transmitter unit local to the receiver unit for configuration of the local transmitter unit for subsequent transmission bursts to a remote receiver unit associated with the remote transmitter unit, thereby to provide open-loop feedback.

15

20

15. A receiver unit according to any one of claims 1 to 12, wherein the signal quality metric is internally used in a blind detection scheme to reconfigure the receiver unit for decoding subsequent transmission bursts.

25

16. A system comprising a receiver unit according to any one of claims 1 to 12 in combination with a transmitter unit, wherein the transmitter unit has an input connected to the output of the receiver unit for receiving the signal quality metric, the transmitter unit having a configuration that is responsive to the signal quality metric.

17. A system according to claim 16, wherein the transmitter unit comprises an interactive multimedia encoder having a configuration that is responsive to the signal quality metric.

30

18. A system according to claim 16 or 17, wherein the transmitter unit comprises a modem having a configuration that is responsive to the signal quality metric.

19. A system according to claim 16, 17 or 18, wherein the transmitter unit comprises a channel encoder having a configuration that is responsive to the signal quality metric.

5

20. A system according to any one of claims 16 to 19, wherein the transmitter unit and receiver unit form a transceiver unit.

21. A system according to any one of claims 16 to 19, wherein the transmitter unit  
10 and receiver unit are remote from each other and form respective parts of separate transceiver units.

15

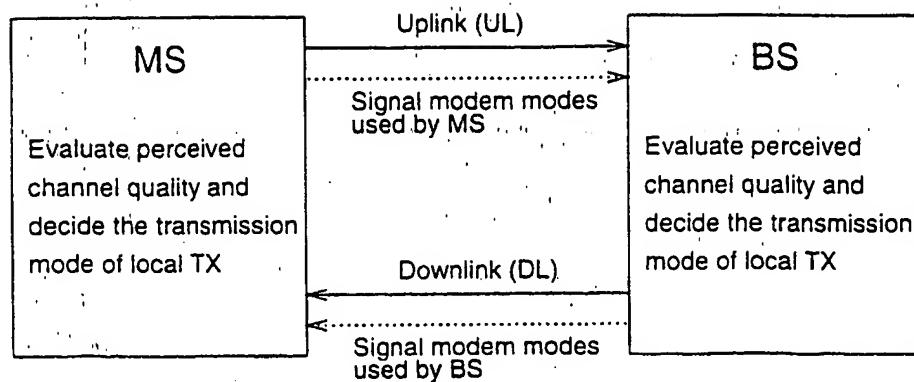


Figure 1(a)

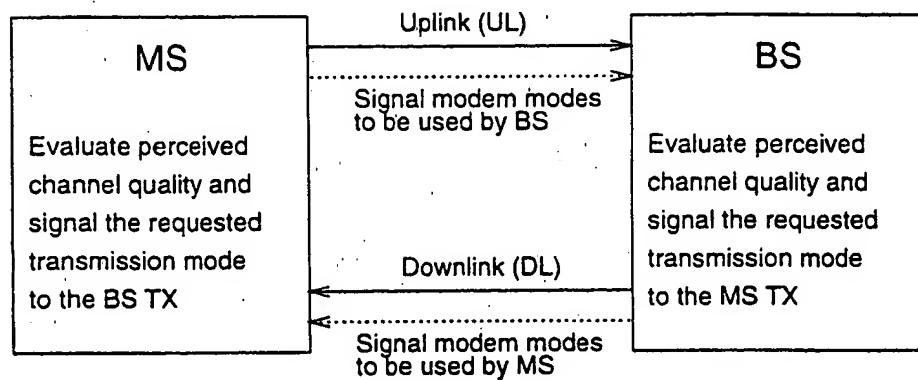


Figure 1(b)

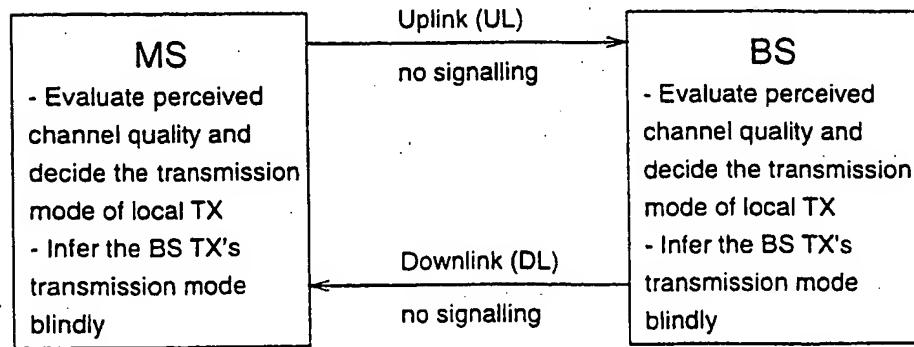


Figure 1(c)

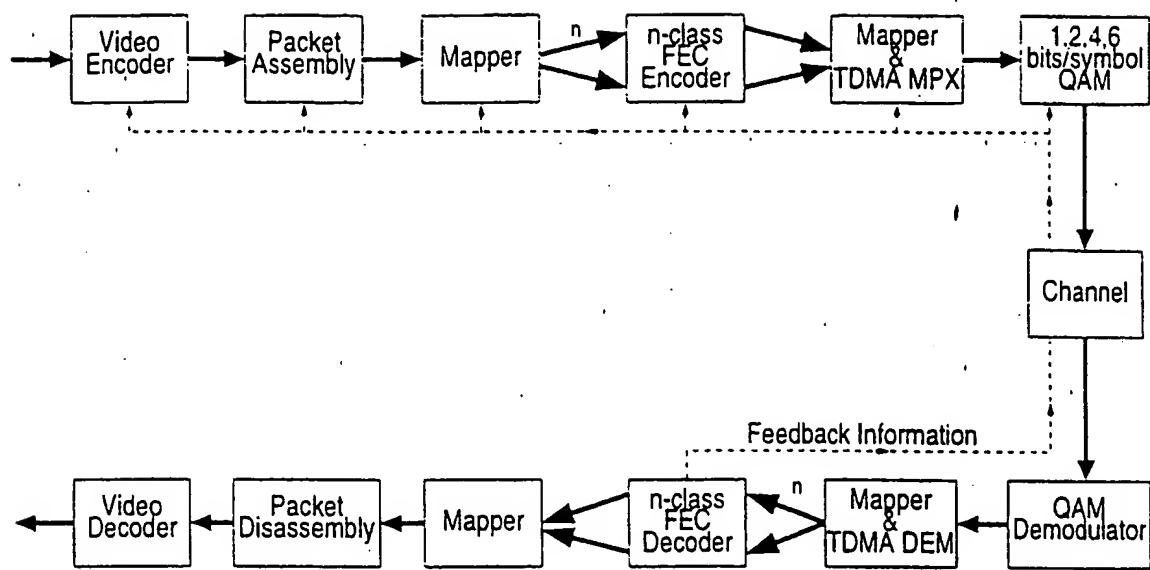


Figure 2

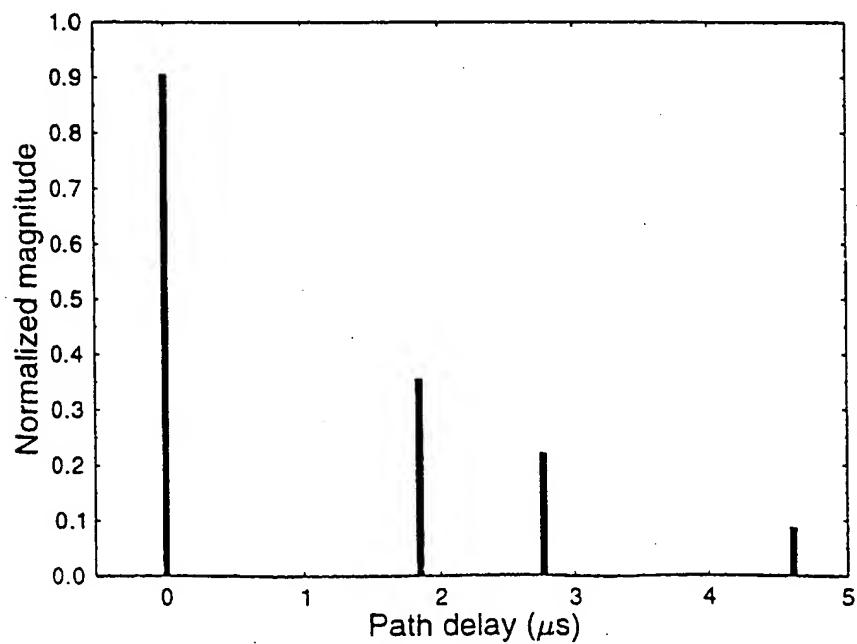


Figure 3

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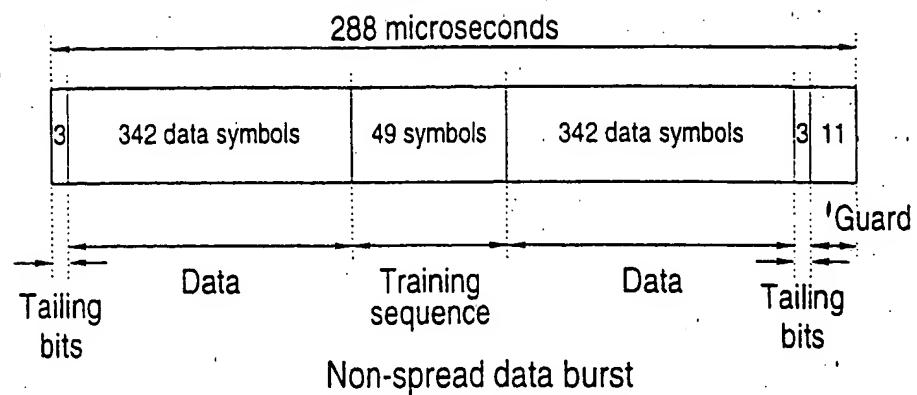


Figure 4

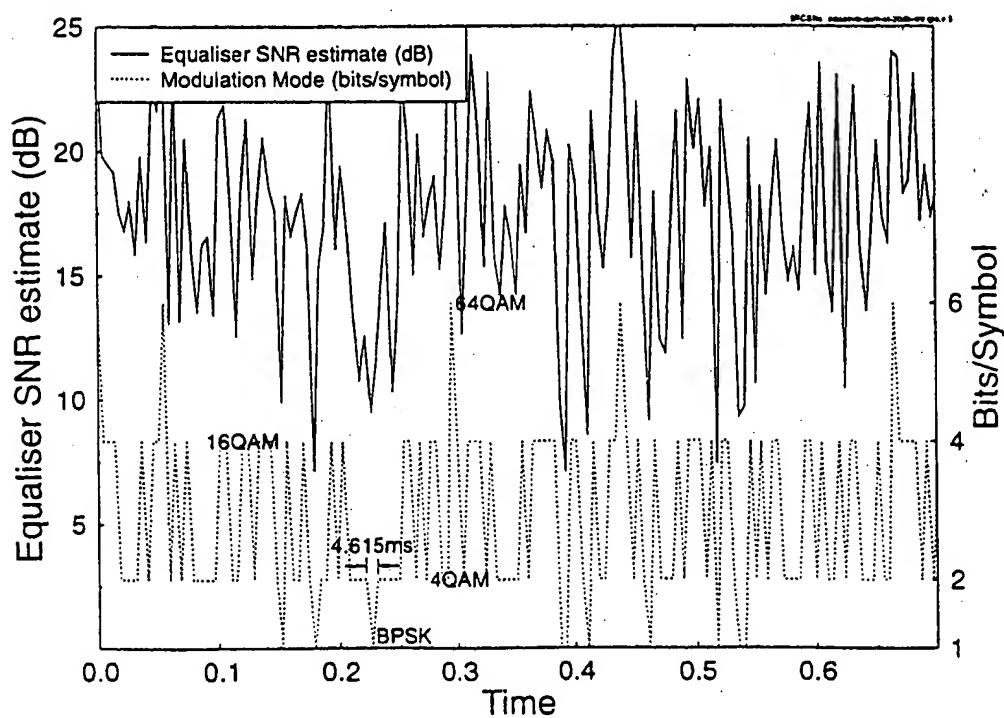


Figure 5

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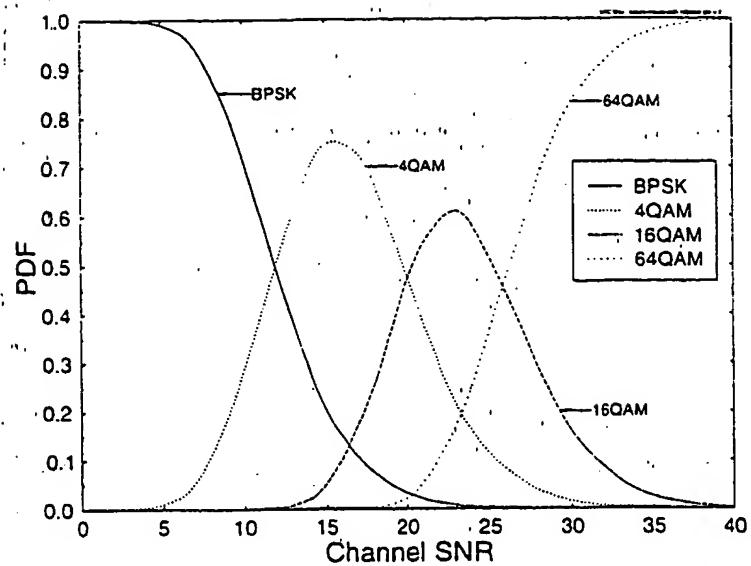


Figure 6

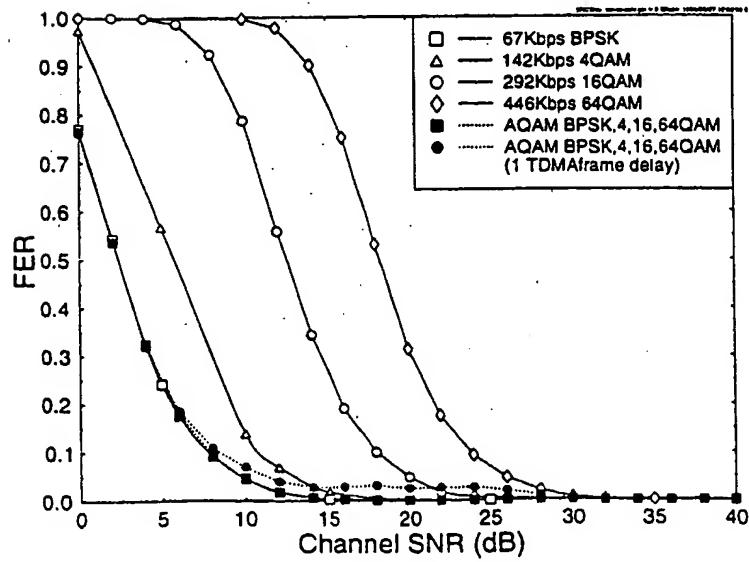


Figure 7

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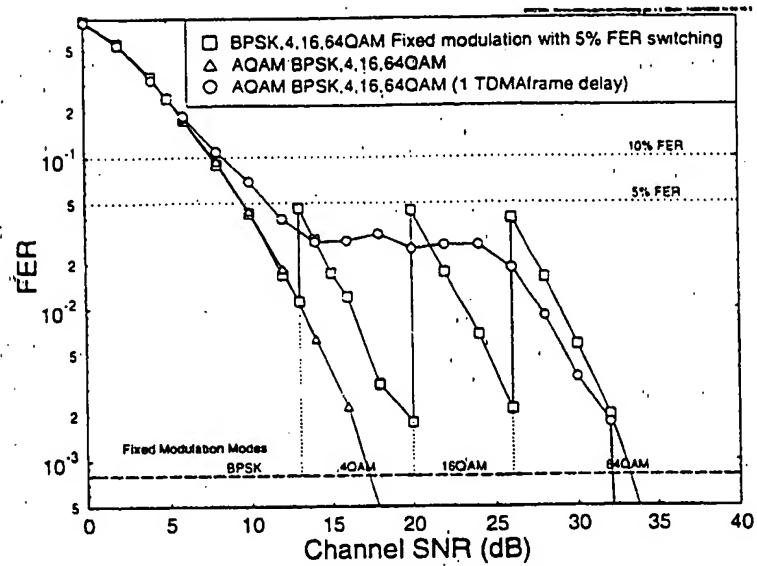


Figure 8

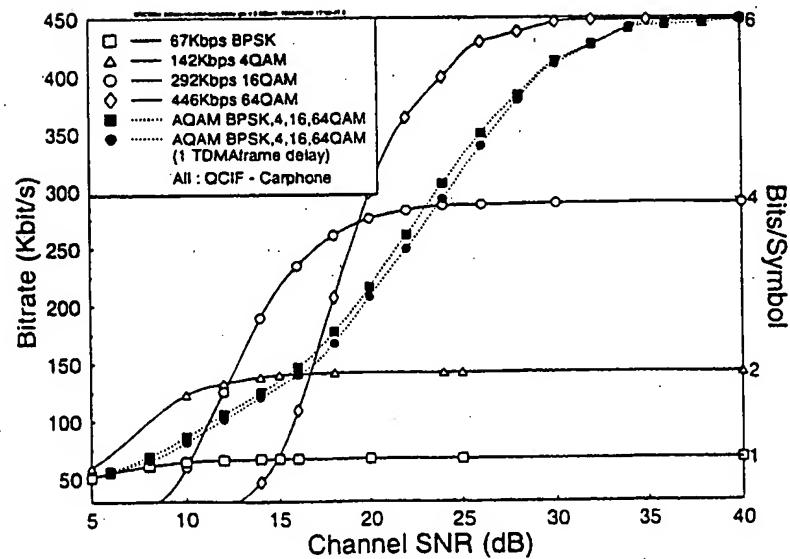


Figure 9

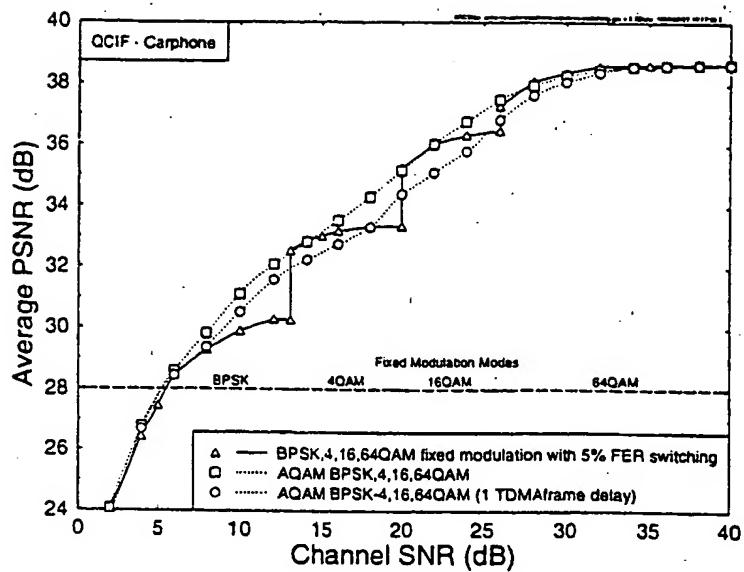


Figure 10

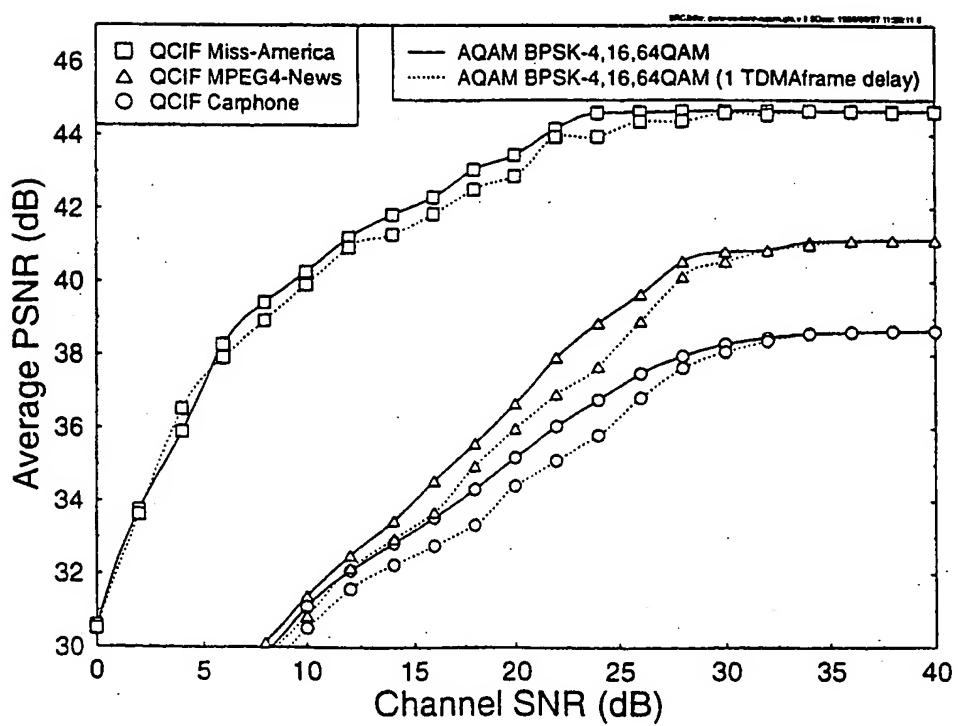


Figure 11

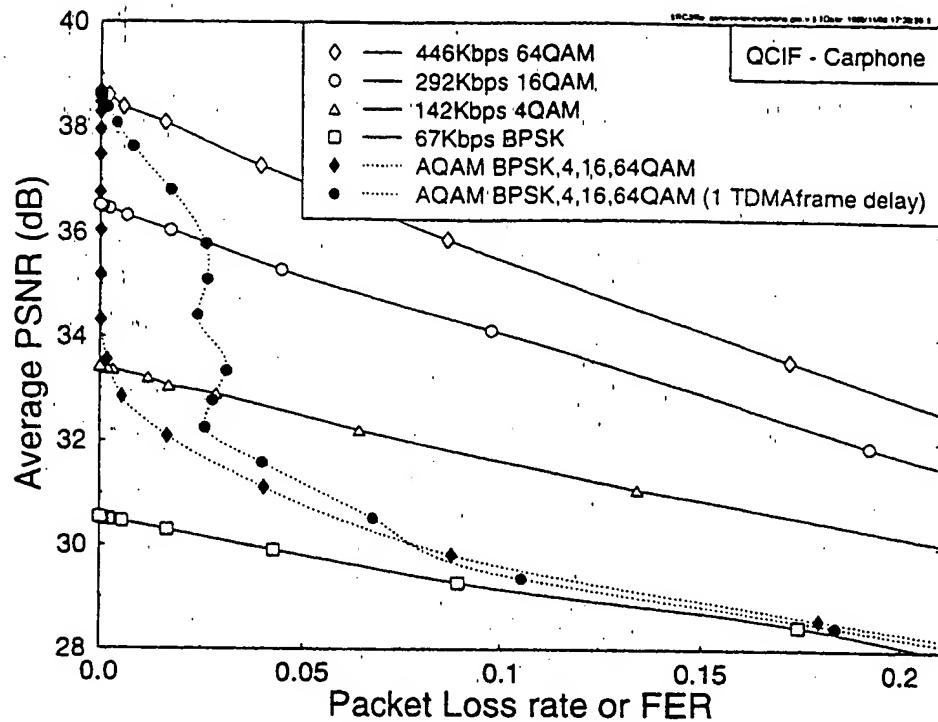


Figure 12

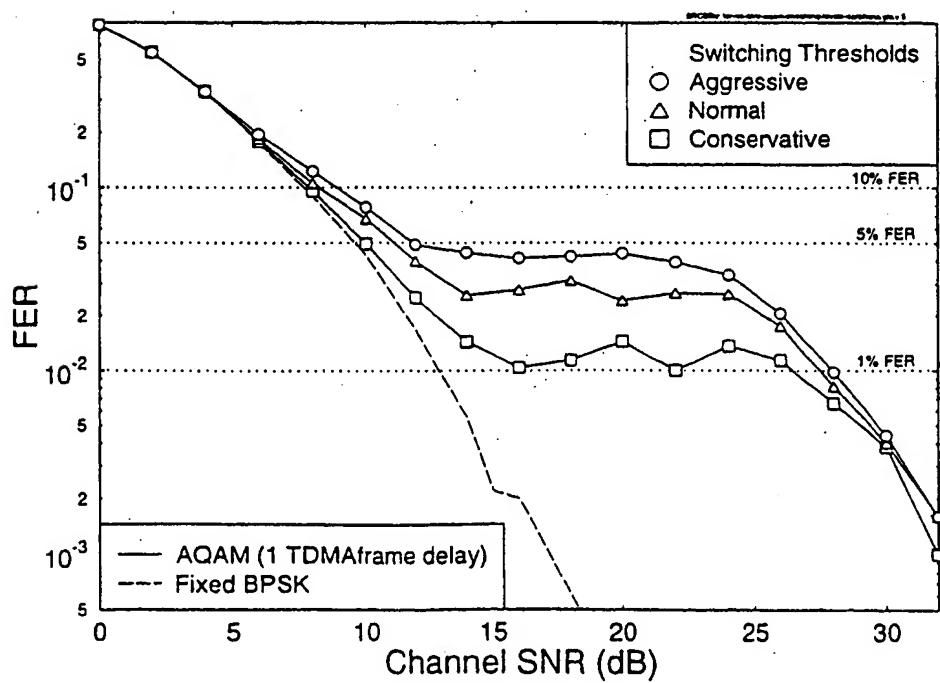


Figure 13

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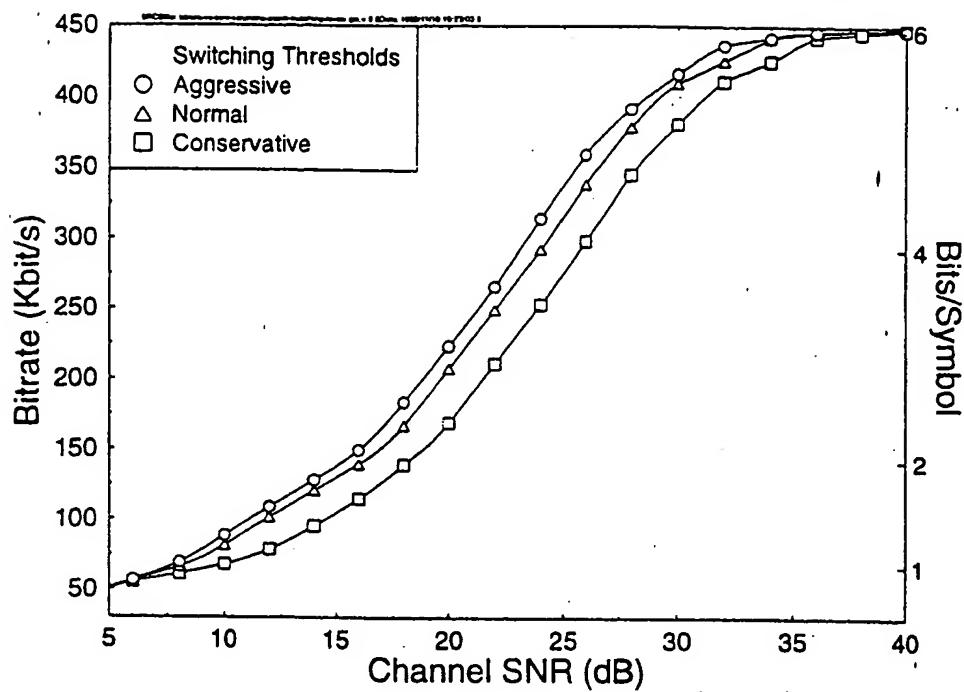


Figure 14

# INTERNATIONAL SEARCH REPORT

Intern. Application No  
PCT/GB 00/00017

**A. CLASSIFICATION OF SUBJECT MATTER**  
IPC 7 H04B1/10

According to International Patent Classification (IPC) or to both national classification and IPC

**B. FIELDS SEARCHED**

Minimum documentation searched (classification system followed by classification symbols)  
IPC 7 H04B

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the International search (name of data base and, where practical, search terms used)

**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	<p>WO 98 51111 A (KONINKL PHILIPS ELECTRONICS NV ;PHILIPS AB (SE)) 12 November 1998 (1998-11-12)</p> <p>page 14, line 30 -page 16, line 31; figures 1,8 claims 6,14 page 5, line 30 -page 6, line 14 abstract</p> <p style="text-align: center;">— —/—</p>	1,7, 12-14, 16-18, 20,21

Further documents are listed in the continuation of box C.

Patent family members are listed in annex.

\* Special categories of cited documents :

- "A" document defining the general state of the art which is not considered to be of particular relevance
- "E" earlier document but published on or after the International filing date
- "L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
- "O" document referring to an oral disclosure, use, exhibition or other means
- "P" document published prior to the International filing date but later than the priority date claimed

"T" later document published after the International filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

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Date of the actual completion of the International search

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## INTERNATIONAL SEARCH REPORT

International Application No  
PCT/GB 00/00017

## C(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	<p>OTSUKI S ET AL: "PERFORMANCE OF MODULATION-LEVEL-CONTROLLED ADAPTIVE MODULATION SYSTEMS" ELECTRONICS &amp; COMMUNICATIONS IN JAPAN, PART I - COMMUNICATIONS, US, SCRIPTA TECHNICA, NEW YORK, vol. 79, no. 7, 1 July 1996 (1996-07-01), pages 81-93, XP000696376 ISSN: 0756-6621 page 82, right-hand column, line 22 -page 84, left-hand column, line 21; figures 1,2</p>	1,16
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